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DIGITAL VIDEO:
reduction of sampling frequency to 11·9 MHz

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Research Department, Engineering Division
THE BRITISH BROADCASTING CORPORATION

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Summary

This report describes some of the problems involved in reducing the sampling frequency in 625-line television digital codecs from 13·3 MHz to 11·9 MHz. Such a reduction would enable a 9-bit coded signal to be transmitted within a serial bit rate of 107 Mb/s, instead of 120 Mb/s which is required for 13·3 MHz sampling.

The sampling frequency of 11·9 MHz is still greater than the Nyquist minimum rate for an analogue video bandwidth of 5·5 MHz, and the main problem is the specification of filters used before sampling in the a.d.c. and after reconstitution of the analogue signal in the d.a.c. It is shown that the post-reconstitution filter demands the more stringent specification, and a filter built to this specification is described. The performance of a codec using the reduced sampling frequency was checked by subjective tests, and found to be satisfactory. However, the considerable group delay variations introduced by the filter in the upper video band may limit the number of codecs which could be cascaded on a transmission link.

1. Introduction

The present BBC experimental digital codec[1] for 625-line System I colour television signals produces, with parity-checking digits, a serial bit rate of about 120 Mb/s. However, practical long-distance television links may handle a rate of only 107 Mb/s. It is therefore desirable to investigate various possibilities for reducing the existing bit rate, without causing an unacceptable degradation of performance.

It is expected that techniques for considerably reducing the bit rate by exploiting various redundancies in the television picture will be devised. Such techniques may require complex instrumentation, and they may not be entirely suitable for all practical applications, at least for a considerable time. An alternative approach, for achieving a moderate reduction in bit rate, is to reduce the frequency at which the television analogue waveform is sampled. This report discusses some of the aspects concerned with using a sampling frequency of 11·9 MHz, which is sufficiently low to enable a 9-bit-per-word digital television signal to be accommodated within a total serial bit rate of 107 Mb/s.

It is assumed that 8 bits per sample are required to produce a picture sufficiently free of quantising noise. In the present equipment the sampling rate is three times the colour subcarrier frequency, i.e., 13·3 MHz, so that 8 bits per sample require a bit rate of 106·4 Mb/s. This is within the desired specification, but if one parity bit is transmitted with each word, for the purpose of error detection, the rate would increase to 119·7 Mb/s. However, the bit rate for 9-bit words could be reduced to 107 Mb/s by sampling at 11·9 MHz, which is still greater than the limiting value of the Nyquist sampling frequency for an analogue signal whose maximum video frequency is 5·5 MHz. In principle, therefore, by sampling at 11·9 MHz it is possible to encode the signal without introducing unacceptable impairment to the television picture, and still allow a parity check bit to be transmitted with each coded word.

The use of 11·9 MHz sampling would not allow advantage to be taken of one technique[2] for digit error detection and concealment, which uses the third previous sample to replace the erroneous sample, both being in nearly the same colour subcarrier phase when the sampling rate is three times subcarrier frequency. Replacement by the previous sample gives somewhat inferior results on a colour picture, but this method of error concealment could be used with the reduced sampling frequency without difficulty. It is considered that the protection afforded may be adequate or that, if necessary, other improved methods of protection could probably be devised.

2. Practical considerations

In practice, the sampling frequency, f_s, must normally be higher than twice the maximum analogue frequency (Nyquist rate), because the sampling process produces 'sidebands' such that each baseband frequency, f, is accompanied by 'image' components at frequencies $(f_s \pm f)$, $(2f_s \pm f)$, $(3f_s \pm f)$, and so on. The $(f_s - f)$ component, often known as the 'alias' component, for a maximum baseband frequency f_m, moves towards f_m as f_s is reduced close to the theoretical minimum value $(2f_m)$ and, because of the beat between components f_m and $(f_s - f_m)$, can introduce interference within the baseband. Undistorted

performance near the minimum sampling rate therefore necessitates accurate low-pass filtering before sampling in the analogue-to-digital converter (a.d.c.), and also after reconstitution in the digital-to-analogue converter (d.a.c.). The filtering is particularly important in the latter case, if the output of the d.a.c. is applied to a device with non-linear characteristics. One example of such a device is a television picture tube. Practical filters are inevitably imperfect, but knowledge of the subjective effects of unwanted components in a television signal enables feasible cut-off and stop-band characteristics of the pre-sampling and post-reconstitution filters to be specified.

2.1. Pre-sampling filtering

Although the maximum video frequency specified in the System I television standard is 5·5 MHz, some energy may be present above this frequency. This may be generated at the picture source, or may be caused by low-level distortion products. In an analogue system these high-frequency components are normally unimportant, because the interference patterns they produce are of such a fine structure as to be practically invisible. However, in a digital system the $(f_s - f)$ alias components resulting from these distortion products may fall below 5·5 MHz and consequently tend to become visible, particularly if they fall in the region of the colour subcarrier frequency. The specification of a filter, required to reduce this effect to a suitably low level, depends upon the shape of the input signal energy spectrum. For the present purpose the signal spectrum is assumed flat, so that the spectrum of the filter output is defined completely by the filter characteristic. This characteristic is chosen so that the alias $(f_s - f)$ components correspond to signals of acceptably low level, even if full-amplitude signals above 5·5 MHz were applied to the filter.

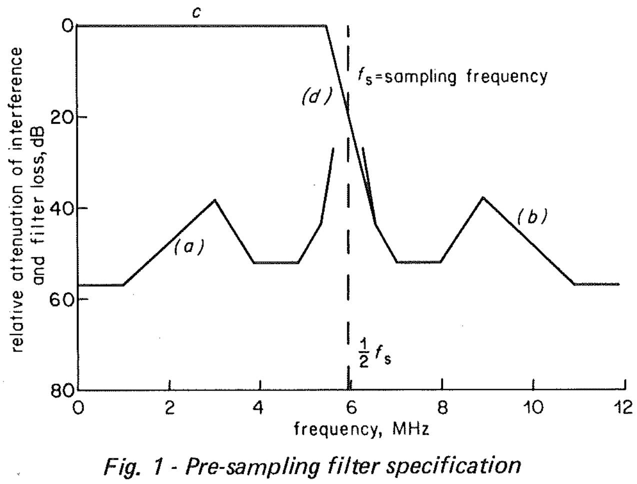

Fig. 1 - Pre-sampling filter specification

(*a*) permissible level of interference in video band
(*b*) components above 6·4 MHz producing (*a*)
(*c*) in-band attenuation
(*d*) out-of-band attenuation immediately above 5·5 MHz

Fig. 1(*a*) shows the estimated permissible level of interfering signals. This is based on a proposed amendment to a CCIR curve giving the protection ratio required against interference from a CW signal to a broadcast television signal.[3] This protection ratio applies when the interference is only present for a small percentage of the time, in which case an impairment in the region of grade 3·5 on the EBU 6-point impairment scale can be accepted. For the present requirement, the interfering signal must be considered as being potentially present all the time, and the impairment must be imperceptible. To take account of this, 15 dB has been added to the CCIR figure for protection ratio. On the other hand, 8 dB has been subtracted, because the protection ratio figures relate to interference between r.f. signals applied to a vestigial-sideband receiver, whereas the present problem is concerned with interference between video signals.[4] The net adjustment therefore is a 7 dB increase in the CCIR protection ratio.

The interfering signals represented by curve (*a*) of Fig. 1 are caused by the presence of the $(f_s - f)$ alias components for $f > 6{\cdot}4$ MHz so the filter out-of-band characteristic required is the mirror image of curve (*a*). This is given by Fig. 1(*b*). Curve (*c*) is the pass-band of the filter, and is taken as flat from 0 to 5·5 MHz. A straight line, (*d*), is then constructed from the 5·5 MHz point such that the attenuation it represents is always equal to or greater than that given by curve (*b*). The connected curves (*b*), (*c*) and (*d*) then define the required filter characteristic.

In this case, the cut-off slope, given by (*d*), is 41 dB per MHz.

2.2. Filtering after digital-to-analogue conversion

After the d.a.c., the reconstituted samples are fed to a low pass filter. Imperfect filtering may cause impairment of the signal, because the reconstituted spectrum contains out-of-band alias frequencies $(f_s - f)$, corresponding mainly to values of f up to 5·5 MHz, although residual higher-frequency components of f may also be significant. These alias components will therefore tend to be of a high frequency, and in a linear system they would have little effect. However, if the signal is applied to a non-linear system, intermodulation products will be generated, and these may cause an additional impairment. One unavoidable and severe source of non-linearity in a television system is the display tube itself, which has a 'gamma' of up to about 2·8.

In order to produce a filter specification which will allow the decoded signal to be applied to a display tube, we will consider the stringent case in which the wanted baseband component is a full-amplitude sine wave of frequency f, and the alias component at $(f_s - f)$, resulting from the sampling process, has a relative amplitude k.

For a gamma of γ the tube brightness is proportional to

$$[1 + \cos(2\pi f\, t) + k \cos(2\pi\, [f_s - f]\,)t]$$

The binomial expansion of this expression shows that in addition to the wanted frequency f and an alias frequency $(f_s - f)$ similar to that occurring in the normal sampling process, there is also an intermodulation product f_i of frequency $(f_s - 2f)$ which can fall within the baseband. The wanted signal amplitude is γ, whereas the amplitude of the intermodulation product is ½k$\gamma(\gamma - 1)$. Hence we find

that if $\gamma = 2{\cdot}8$,

$$\frac{\text{intermodulation signal, } (f_s - 2f)}{\text{wanted signal, } (f)} \simeq k$$

so that the amplitude of the intermodulation product is nearly the same as that of the alias component.

The stopband characteristic of the post-reconstitution filter can therefore be derived as follows. The intermodulation product $f_i = f_s - 2f$ arises from the alias frequency $f' = f_s - f$. To establish the filter characteristic required for f' we observe that

$$f' = \tfrac{1}{2}(f_i + f_s)$$

Therefore, since the permissible level of f_i is given by Fig. 1, curve (*a*), the curve defining the permissible level of f' is similar to this curve but with f_i replaced by $\tfrac{1}{2}(f_i + f_s)$. The resulting characteristic is shown in Fig. 2, for a sampling frequency of 11·9 MHz. The slope of the cut-off is chosen to ensure that interference up to 5·5 MHz is within the prescribed limit, by intersecting the stopband characteristic at 6·4 MHz, i.e. at $f_s - 5{\cdot}5$ MHz. The cut-off rate for this filter is 62 dB per MHz, which is a high slope for a practical filter.

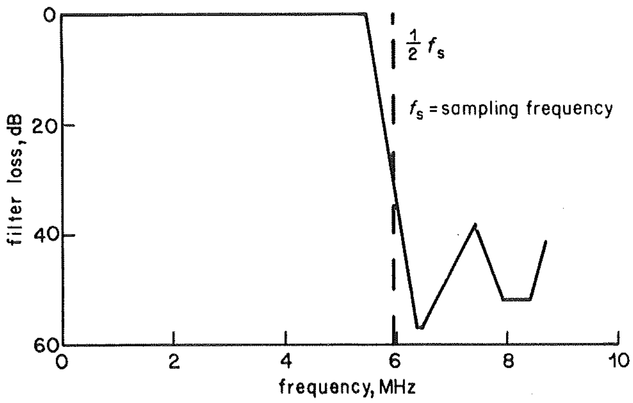

Fig. 2 - Post-reconstitution filter specification

2.3. The use of sampling frequencies which are odd multiples of half line frequency

A post-reconstitution filter with a characteristic approximating to the specification developed above will be described in Section 3.2. It is, however, convenient at this point to examine the technique of sampling at a frequency which is an odd multiple of half line frequency, since this may allow some relaxation in the specification for the pre-sampling and post-reconstitution filters.

Except for the part of the spectrum in the region of the colour subcarrier, most of the energy in a video waveform is distributed at or near spectral lines which are multiples of line frequency. Interference occurring at odd multiples of half line frequency is less objectionable than interference at other frequencies, particularly line-frequency multiples.

By a suitable choice of sampling frequency it is possible to ensure that the alias components at $(f_s - f)$ and the intermodulation products at $(f_s - 2f)$ occur at odd half-line-frequency multiples when the baseband frequency f is a line-frequency multiple.

Let the baseband frequency be mf_1 where m is an integer and f_1 is the line frequency. For the alias component to be an odd half line frequency multiple, say $(n + \tfrac{1}{2})f_1$, n being an integer, it follows that

$$f_s = mf_1 - (n + \tfrac{1}{2})f_1,$$

so that f_s must be an odd multiple of half line frequency.

Fig. 3(*a*) shows the estimated permissible level of interference when the interfering signal spectrum is offset (with an accuracy of ±500 Hz) from the wanted signal spectrum by an odd multiple of half line frequency.[3] Comparison with the corresponding characteristic in Fig. 1(*a*) shows that a considerable increase in the level of the interfering signal can be tolerated. Figs. 3(*b*) and (*c*) show the specifications of the pre-sampling and post-reconstitution filters for 11·9 MHz sampling, based on Fig. 3(*a*). It will be seen that both cut-off characteristics can be made less sharp, the post-reconstitution rate of cut-off being reduced to 43 dB per MHz.

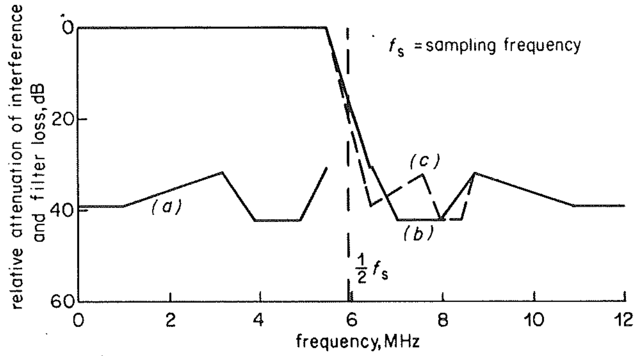

Fig. 3 - Filter specifications for sampling at an odd multiple of half line frequency

(*a*) permissible level of interference
(*b*) ———— pre-sampling filter
(*c*) – – – – post-reconstitution filter

3. Experimental investigations

3.1. Preliminary experiments

Experiments were carried out to check the validity of the preceding discussion. Attention was paid primarily to the specification to the post-reconstitution filter, because the conclusions reached in Section 2 indicate that this is likely to be more stringent. The pre-sampling filter used in the experiments had a cut-off slope of 16 dB per MHz. The output of the digital codec was observed on monochrome and colour monitors, while the sampling frequency was controlled by an external oscillator. Initially the 'picture' information consisted of a sine wave excursing from white to just above black level. In subsequent experiments test cards, colour bars, slides and off-air programmes were used.

Tests were first carried out using a post-reconstitution filter having a cut-off slope of 16 dB per MHz. The sine wave video signal frequency was adjusted to 5·500 MHz (a multiple of line-frequency), so that the picture appeared as stationary vertical bars. The sampling frequency was then reduced from 13·3 MHz (three times subcarrier) until an interference pattern was discernible on the screen. This occurred at about 12·9 MHz. Using the predictions in Section 2.2, it may be expected that interference should be visible for sampling frequencies up to 13·7 MHz. That no interference was noticed at sampling frequencies above about 12.9 MHz probably indicates that the monitor frequency characteristics caused the cut-off to be steeper than that of the filter alone.

From the point of view of interference from alias and intermodulation components, a 5·500 MHz full-amplitude sine wave is the most stringent 'picture' signal likely to be encountered. In similar tests with colour-bars and average programme material and still using the filter with a cut-off rate of 16 dB per MHz, the sampling frequency could be reduced to 11·0 MHz without visible impairment. Chrominance beat patterns were observed as the sampling frequency was reduced below 10·9 MHz; these were due to intermodulation products falling within the chrominance channel.

The use of sampling at an odd multiple of half line frequency was investigated briefly. It was found that in this case negligible impairment of the sine wave 'picture' occurred at sampling frequencies down to about 12·4 MHz. This is consistent with the corresponding theoretical figure, 12·5 MHz, derived in a similar way to that shown in Fig. 2. A further decrease would be permissible with average programme material.

The sharpest-cut video filter available at the time* had a cut-off slope of about 26 dB/MHz, and when inserted in tandem with the existing filter in the digital equipment, increased the total post-reconstitution filter cut-off slope to 42 dB/MHz. According to the preceding discussion, visible interference should not then occur until the sampling frequency is reduced to 11·9 MHz, provided that the frequency is adjusted to be an odd multiple of half line frequency, as described in Section 2.3. Limited subjective tests confirmed this.

A further experiment was conducted to check the proposed specification of the pre-sampling filter. For this test the input signal was a full-amplitude sine wave of frequency f, greater than 5·5 MHz, and the pre-sampling filter in the equipment was by-passed. In this condition the alias component at $(f_s - f)$ is generated at almost full amplitude, with only a small reduction from the maximum arising from instrumental imperfections. The two signals at frequencies f and $(f_s - f)$ appear at the post-reconstitution filter, which in this case was the original one having a cut-off slope of 16 dB per MHz. Since f lies in the stop-band of this filter, its amplitude was substantially reduced, and only the alias component $(f_s - f)$ due to the sampling process in the a.d.c. appeared on the monitors. The input signal f was adjusted in amplitude until the $(f_s - f)$ pattern was just not visible.

* BBC type FL4/556A.

The results of this test, conducted with a sampling frequency of 11·9 MHz, are shown in Fig. 4. The cut-off slope specified in Section 2.1 appears from this test to be somewhat too stringent, but unless there are significant practical advantages in slackening its specification, it forms a reasonable objective for the pre-sampling filter characteristic.

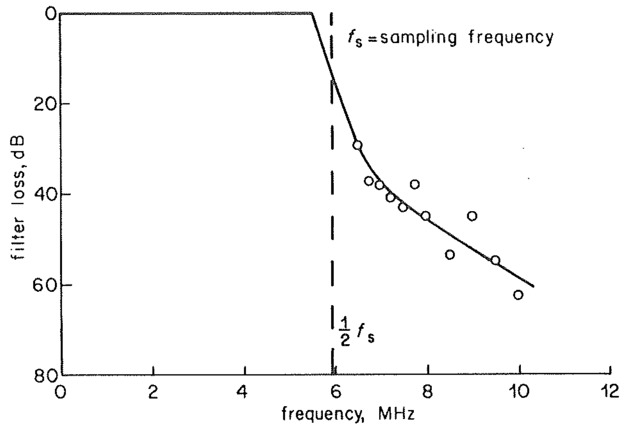

Fig. 4 - Pre-sampling filter specification, deduced from subjective tests
o Points obtained from tests

3.2. A post-reconstitution filter for 11·9 MHz sampling

As discussed in Section 2.2, this filter is required to meet a more stringent specification then the pre-sampling filter, and it is important to know whether a practical realisation of such a filter is feasible. A design was accordingly undertaken,* and the resultant circuit is given in Fig. 5. It comprises the basic low-pass sections, which provide the required amplitude characteristic, four group-delay correction sections, and a gain equaliser section which was added to correct for the fall in response at high frequencies caused by the group-delay equaliser, and the sample-and-hold circuits in the codec. The low-pass sections are m-derived networks, in which the frequencies of infinite attenuation are chosen to take advantage of the fact that the insertion loss can be allowed to rise in the region of 7·5 MHz (see Fig. 2). No pre-correction techniques were employed to take account of coil losses, so some degradation in performance may reasonably be expected, but this was calculated to be small with coils having a Q-factor in the region of 100. The parameters of the group-delay correction sections were optimised to ensure that the total delay at the colour subcarrier frequency equalled that at zero frequency, and also that the magnitude of the ripples in the delay frequency characteristic increased smoothly with frequency.

Measured amplitude and group delay responses of the digital equipment including this filter, and with a sampling frequency of 11·9 MHz, are shown in Fig. 6. It is seen that a good performance is obtained, apart from a loss of 3 dB at 5·5 MHz. This is considered satisfactory, since it is reasonable to suppose that an improved performance could

* This work was carried out by BBC Designs Department.

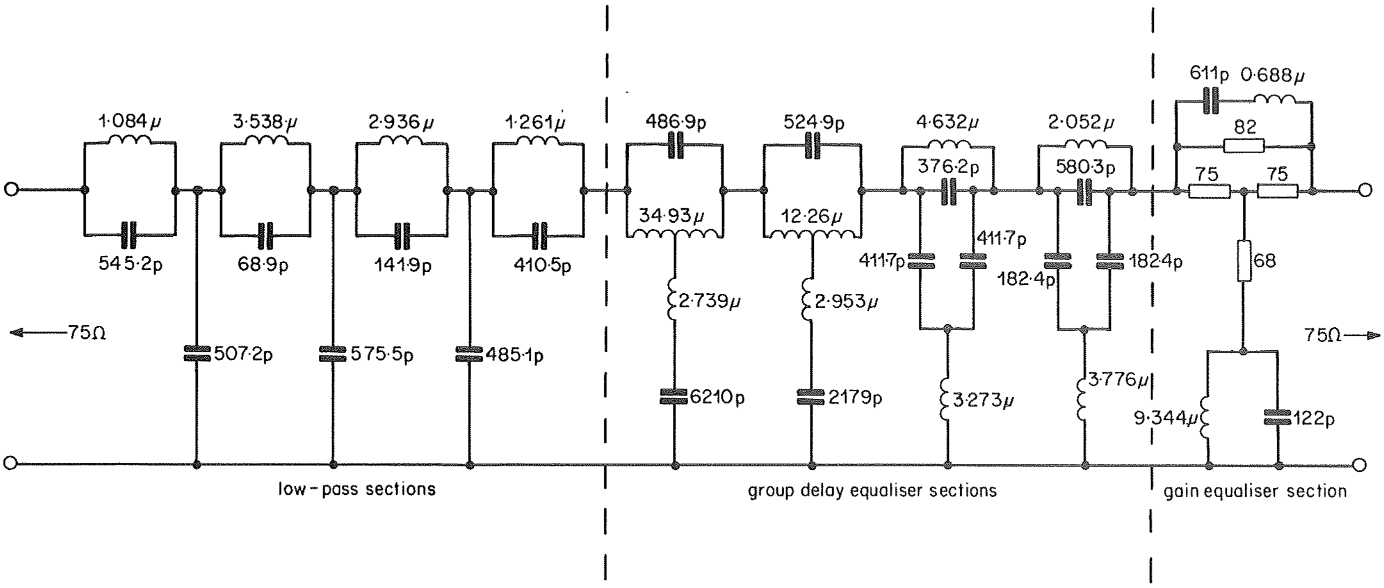

Fig. 5 - Circuit of post-reconstitution filter

be obtained if pre-correction were applied in the initial filter design, to take account of coil losses. Photographs of the responses of the codec, including the new filter, to 2T and 10T pulses are given in Fig. 7. The response to the 10T chrominance pulse is of particular interest because it confirms that the chrominance-luminance delay inequality is small, in spite of the large variations in group delay within the chrominance band.

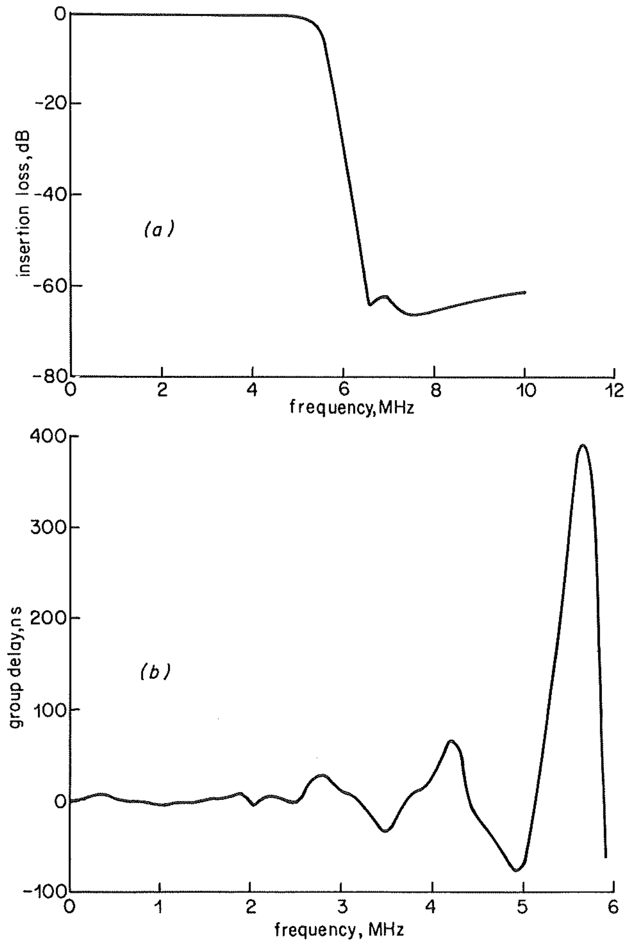

Fig. 6 - Response of codec including new post-reconstitution filter, and sampling at 11·9 MHz
(*a*) Amplitude (*b*) Group delay

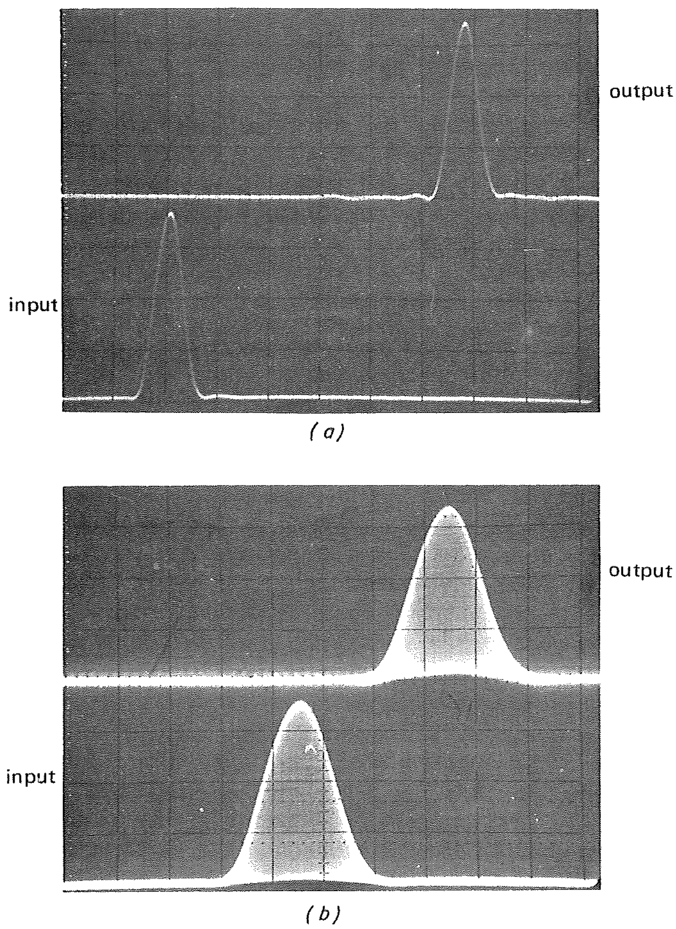

Fig. 7 - Pulse response of codec including new post-reconstitution filter, and sampling at 11·9 MHz
(*a*) 2T luminance pulse (*b*) 10T chrominance/luminance pulse

Examination of several critical still pictures containing highly-saturated vertical coloured detail, and a number of typical moving pictures, showed that the filter introduces no perceptible additional impairment. With a full amplitude 5·5 MHz sine-wave 'picture' signal, the sampling frequency could be reduced to 11·85 MHz before a beat pattern became discernible.

Thus, the experiments showed that it is possible to realise a practical filter which would permit a sampling frequency of 11·9 MHz to be used, at least in the case where only one codec is required. However, in order to assist in the design of such filters, and also to provide information on the possible performance if more than one codec is employed in a transmission system, it is desirable to specify the permissible variations of group delay in the upper part of the video band. This was the subject of a separate investigation,[5] which resulted in the tentative specification shown in Fig. 8. This gives the limits corresponding to a mean impairment grade of 1·5 on the EBU 6-point scale, i.e. at least 50% of viewers would notice no impairment, while most of the remaining viewers would rate the impairment as 'just perceptible'. Also illustrated in Fig. 8 are the delay of a single codec using the filter described above, and the estimated total delay of three similar cascaded codecs. The curves suggest that three codecs could be accommodated if the only source of group delay variations on a transmission link arose in these codecs, and if grade 1·5 impairment is taken as the acceptance limit.

In practice, it may be unnecessary to include the sharp-cut reconstitution filter at every codec in a transmission link, because the specification of this filter is associated with the non-linearity of the display tube rather than with the sampling and digital coding processes. The group delay response of the pre-sampled filter would then become a more significant part of the total delay, but its group delay response could be made flatter than that of the sharp-cut post-reconstitution filter, and more codecs could be cascaded.

4. Alternatives to analogue filters

It has been shown that practical analogue filters can be constructed for a codec sampling at 11·9 MHz, but the large group delay introduced at the upper end of the video band by the post-reconstitution filter would limit the number of codecs which could be used in tandem. It is therefore worthwhile considering possible alternatives to conventional filters. Examples are comb filters and the use of digital filtering techniques.

A comb filter is a filter having a periodic amplitude-frequency response. In the present application the pass-bands or 'teeth' would be situated at line-frequency multiples and the filtering action would be confined to the upper part of the video band. The purpose of the comb filtering technique, which has been used in sub-Nyquist sampling of the luminance signal,[6] is to ensure that the disturbing aliasing or intermodulation components of the interfering signal spectrum lie between the teeth of the comb. Therefore, in the present application, where most

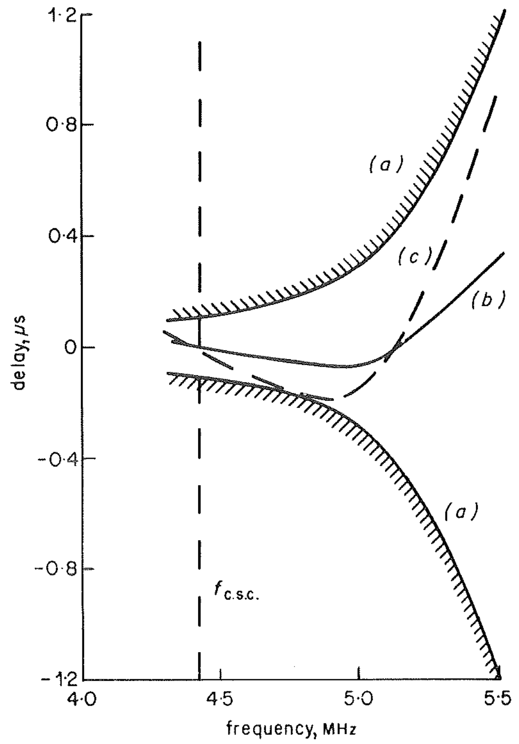

Fig. 8 - Group delay characteristic of one and of three codecs, compared with permissible limits for grade 1·5 impairment

(*a*) permissible limits (*b*) one codec (*c*) three codecs

of the wanted signal energy is grouped around multiples of line frequency, the sampling frequency would be adjusted so that the intermodulation products lie at odd multiples of half line frequency. This is a similar condition to that discussed in Section 2.3, where the sampling frequency is an odd multiple of half line frequency. The comb filter would of course eliminate baseband components at odd half-line-frequency multiples, and thereby reduce the definition of diagonal edges.

In the case of colour systems, further impairments may occur. With the NTSC system the colour information would be largely removed by a comb filter extending through the subcarrier region. In the case of PAL signals considerable picture impairment can occur. In the present application, the impairment inevitably introduced by comb filtering would probably be greater than the reduction in interference stemming from the high-frequency alias components.

Digital filters,[8] simulating the characteristics of analogue low-pass filters, appear to offer advantages in providing the possibility of a greater degree of control over the group delay response, but they would require a large number of delay and arithmetic elements. In recursive digital filters each output sample is formed from previous output and input samples. This type of filter requires considerably fewer delay and arithmetic elements than the non-recursive type which processes only input samples. As a rough estimate, about ten delays, multiplications and additions would be required in a recursive filter for post-reconstitution filtering. Binary arithmetic introduces such

instrumental complexity, however, that at the present sampling rate, a digital filter would be impracticable. However, digital filtering techniques for this application may become practicable in the future as the speed of digital integrated circuits increases and their cost decreases.

5. Conclusions

It is practicable to accommodate a 9-bit-per-sample digital composite video signal within a bit rate of 107 Mb/s, by using a sampling frequency of 11·9 MHz. The use of this sampling frequency imposes relatively stringent requirements on the pre-sampling and post-reconstitution filters. Specifications for both filters have been deduced from consideration of permissible levels of in-band interference, and it was found that the more severe specification is that of the post-reconstitution filter, in the digital-to-analogue converter. This arises from the inherent non-linearity of the television display tube. Experiments largely supported the theoretical approach.

A post-reconstitution filter having the required specification was constructed, and its performance in a single digital codec with a sampling frequency of 11·9 MHz was found to be adequate. Nevertheless the considerable group delay introduced by the filter at the upper end of the video band would limit the number of codecs which could be cascaded. However, it may be unnecessary to include such a sharp-cut post-reconstitution filter in every codec in a transmission link because the specification of this filter is associated with the non-linearity of the display tube. In this case more intermediate codecs could be cascaded in a signal-processing and transmission system.

A worthwhile alleviation of the filter specification could be gained by sampling at an odd multiple of half line frequency.

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